Apparatus for synthesizing analog signals in PCM.

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Abstract

The present invention provides an analog signal synthesizing apparatus including a waveform memory for storing a plurality of analog signals as PCM data sampled with different sampling frequencies, the amount of PCM data corresponding to plural channels being read from the waveform memory and used to synthesize the analog signals. The analog signal synthesizing apparatus includes an oversampling device for shifting the sampling frequency of said PCM data read from said waveform memory for each channel toward the side of high frequency; a summing device for summing the oversampled PCM data for the respective channel; a D/A converter for converting the summed data into an analog signal; and a low-pass filter for setting a cut-off frequency based on the sampling frequency shifted to the side of high frequency and for eliminating aliasing noises included in the PCM data from the synthesized analog signal. The elimination of the aliasing noise included in the PCM data for each channel is carried out at the common low-pass filter.

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(54) **Apparatus for synthesizing analog signals in PCM.**

(57) The present invention provides an analog signal synthesizing apparatus including a waveform memory for storing a plurality of analog signals as PCM data sampled with different sampling frequencies, the amount of PCM data corresponding to plural channels being read from the waveform memory and used to synthesize the analog signals. The analog signal synthesizing apparatus includes an oversampling device for shifting the sampling frequency of said PCM data read from said waveform memory for each channel toward the side of high frequency; a summing device for summing the oversampled PCM data for the respective channel; a D/A converter for converting the summed data into an analog signal; and a low-pass filter for setting a cut-off frequency based on the sampling frequency shifted to the side of high frequency and for eliminating aliasing noises included in the PCM data from the synthesized analog signal. The elimination of the aliasing noise included in the PCM data for each channel is carried out at the common low-pass filter.

FIG. 1A

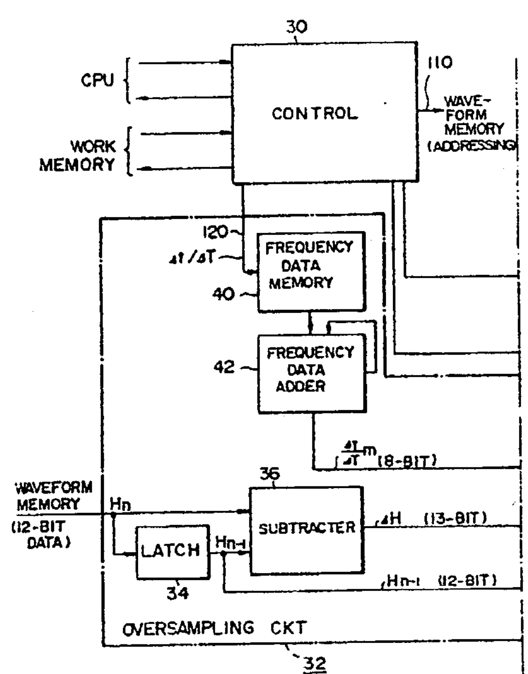
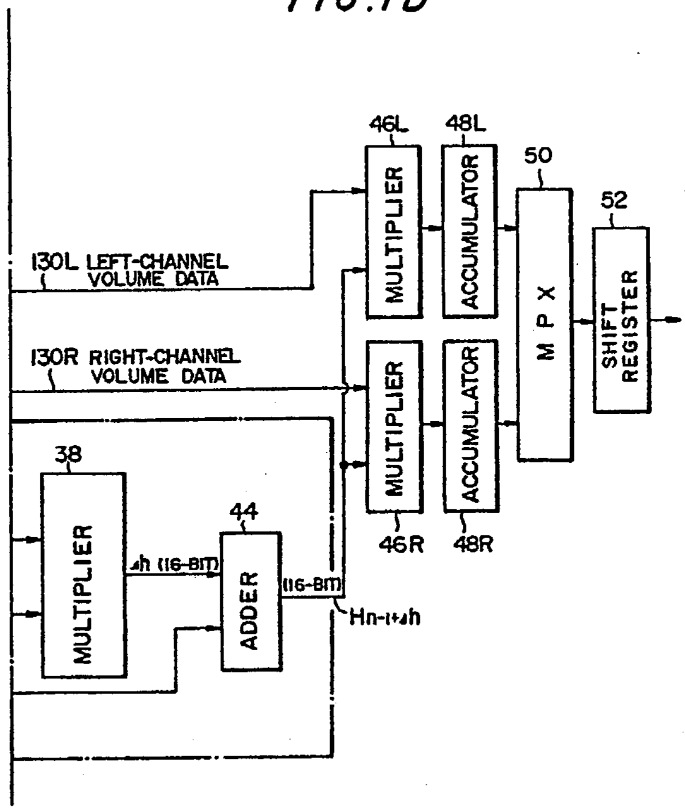


FIG.1B


46L
48L
50
52
MULTIPLIER
ACCUMULATOR
M P X
SHIFT REGISTER
130L LEFT-CHANNEL VOLUME DATA
130R RIGHT-CHANNEL VOLUME DATA
MULTIPLIER
ACCUMULATOR
46R
48R
38
MULTIPLIER
44
ADDER
Δh (16-BIT)
(16-BIT)
Hn-1+Δh

ANALOG SIGNAL SYNTHESIZER IN PCM

BACKGROUND OF THE INVENTION

Field of the Invention:

The present invention relates to an analog signal synthesizing system in PCM and more particularly to such a system capable of reading PCM data corresponding to plural channels from a waveform memory and synthesizing an analog signal from such PCM data.

Description of the Prior Art:

PCM has various superior characteristics such as very high resistance to noise, very high resistance to interference with the adjacent channel and others. Therefore, PCM has been currently utilized in various broadly widened applications such as synthesizers, musical compact disc devices, PCM communication systems and others.

Figure 8 shows the principle of PCM wherein it is supposed that an analog signal such as voice is converted into PCM signal. As shown in Figure 8A, an analog signal is sampled with a predetermined sampling frequency to form a PAM wave as shown in Figure 8B. Such a PAM wave is quantized and encoded to provide PCM data.

If the PCM data so obtained are desired to reconvert into an analog signal, they are first decoded to form the PAM wave as shown in Figure 8B. The PAM wave is then passed through a low-pass filter to reproduce a signal wave having the original signal waveform.

When the analog signal is sampled with the frequency f_s, there is obtained such a waveform spectra as shown in Figure 9. In Figure 9, the hatched part is a spectra in the original analog signal. If such a sampling is carried out, there are created a plurality of aliasing noises at locations integer times the sampling frequency, that is, f_s, $2f_s$, $3f_s$ and so on. These aliasing noises will be superposed over the waveform spectra in the present signal if the sampling frequency f_s is too low. Thus, it becomes impossible to faithfully reproduce the original analog signal.

If the sampling frequency f_s is too high, however, the amount of data to be handled increases to make the data processing very cumbersome.

In order to faithfully reproduce the original analog signal while minimizing the amount of data to be handled, thus, it is necessary to set the sampling frequency as low as possible within a range over which the aliasing noises are not mixed with the original signal.

In accordance with the sampling theorem, the sampling frequency may be set at a level two times or more the maximum frequency in the original analog signal to prevent the waveform spectra thereof from being mixed with the aliasing noises. If the sampling frequency is set at a level two times the maximum frequency of an objective analog signal, the amount of data to be handled can be minimized so that the original analog signal can be faithfully reproduced.

Figure 10 illustrates an example of the conventional analog signal synthesizing systems utilizing such PCM technique.

Such a system comprises a waveform memory 10 adapted to store a plurality of analog signals as PCM data which have been sampled with different sampling frequencies. PCM data corresponding to three channels are read out from the waveform memory 10 to synthesize an analog signal.

For example, if the analog signal synthesizing system is to be used to produce a synthesized sound combining a plurality of musical instruments, voice analog signals from the objective musical instruments, for example, guitar, drums and bass are previously stored in the waveform memory 10 as PCM data which are sampled with frequencies f_{s1}, f_{s2} and f_{s3} corresponding to those of the musical instrument.

PCM data having the respective sampling frequencies f_{s1}, f_{s2} and f_{s3} are read from the waveform memory 10 through the first, second and third channels and converted into analog signals through D/A converters 12-1, 12-2 and 12-3. These analog signals are then inputted into low-pass filters 14-1, 14-2 and 14-3, respectively. Aliasing noises are removed from the inputted analog signals by the respective low-pass filters 14-1, 14-2 and 14-3. Thereafter, the analog signals are applied to a mixer 18 through amplifiers 16-1, 16-2 and 16-3, respectively. At the mixer 18, the analog signals inputted thereinto and corresponding to three channels are mixed to form a synthesized analog from the three analog signals, for example, a synthesized analog sound waveform consisting of the sound waves representative of the guitar, drums and bass.

In order to use the low-pass filter 14 to remove the aliasing noises included in the PCM data, it is necessary to set the cut-off frequencies f_c in each of the low-pass filters 14-1, 14-2 and 14-3 at a level one-half the sampling frequency f_s as shown in Figure 11A. This is because if the cut-off frequency f_c is higher than 1/2 f_s as shown in Figure 11B, a

part of the aliasing noises remains in the PCM data and will be reproduced as noises.

As described hereinbefore, however, the PCM data read from the waveform memory through the first, second and third channels are different from one another in the sampling frequencies f_{s1}, f_{s2} and f_{s3}. There is thus a problem that the low-pass filters 14-1, 14-2 and 14-3 corresponding to the respective channels must be set at different cut-off frequencies f_{c1}, f_{c2} and f_{c3}.

In particular, such a conventional system is designed such that the sampling frequency of the PCM data read out from the waveform memory 10 is in one-to-one relationship with the cut-off frequency f_c in the low-pass filter 14. Therefore, each of the channels is poor in universality. This takes place a problem in that the particular PCM data can be read out only from the corresponding channel, for example, the sound of a guitar from the first channel, the sound of drums from the second channel and the sound of a bass from the third channel.

When it is wanted to read many different PCM data, for example, ten or twenty different PCM data from the waveform memory 10, the number of channels corresponding to the number of the different PCM data portions must be provided. This results in a further problem in that the entire construction of the system becomes costly and more complicated.

SUMMARY OF THE INVENTION

In view of the above problems of the prior art, it is an object of the present invention to provide an analog signal synthesizing system in PCM wherein the aliasing noises included in the PCM data for the respective channels can be removed through a common low-pass filter.

To this end, the present invention provides an analog signal synthesizing system in PCM, comprising a waveform memory for storing a plurality of analog signals as PCM data which are respectively sampled with different sampling frequencies, the PCM data corresponding to plural channels being read out from said waveform memory and used to synthesize an analog signal, the improvement being characterized in that said system comprises:
oversampling means for shifting the sampling frequency of the PCM data corresponding to each of the channels which is read out from the waveform memory toward the side of high frequency;
adding means for summing the oversampled PCM data for each channel;
D/A converting means for converting the summed data into an analog signal; and
a common low-pass filter having a cut-off frequency set based on the sampling frequency shifted toward the high frequency side, said common low-pass filter being adapted to remove aliasing noises included in the PCM data from said synthesized analog signal.

In such an arrangement, the waveform memory stores a plurality of analog signals as PCM data which are sampled with different sampling frequencies.

When the PCM data corresponding to each of the channels are read out from the waveform memory, they are then oversampled. Such an oversampling process may be a process for determining new data utilizing, for example, various interpolations such as primary interpolation, secondary interpolation and so on. There is also a process of using the same data as those of the waveform memory for each oversampled point and digital filtering them as original data to determine new data. Such digital filtering processes include a filtering in the frequency region due to the discrete Fourier transformation (digital low-pass filter) or another filtering in time region fold in the impulse response of the filter (smoothing). The digital filter is more fully described in "Interface", November 1987, No. 126.

On such an oversampling, the read-out PCM data corresponding to each of the channels will have its sampling frequency which is shifted toward the side of high frequency.

It is now assumed that PCM data are read out from the waveform memory through each of the channels as shown in Figure 2A. When the PCM data so read out are oversampled as described, the sampling frequency f_{s1}, f_{s2} or f_{s3} corresponding to each of the channels will be shifted toward a higher frequency f_{DA1}, f_{DA2} or f_{DA3}. In such a manner, the present invention shifts any aliasing noise having its lower frequency included in the PCM data toward the region of higher frequency so that the spectra in the original signal as shown by hatching in Figure 2A are more broadly spaced away from the spectra of the adjacent aliasing noise.

The oversampled PCM data for each channel are summed by adding means and converted into an analog signal through D/A converting means. Thereafter, the analog signal is applied to the low-pass filter. As described hereinbefore, the low-pass filter has its cut-off frequency set based on the sampling frequency shifted toward the higher frequency by the oversampling process, thereby removing any aliasing noise included in the PCM data from the input synthesized analog signal.

In accordance with the present invention, thus, the PCM data corresponding to each of the channels are oversampled to focedly shift the aliasing

noises having their lower frequencies toward the region of higher frequency. Therefore, the PCM data for each channel will have a widened spacing of frequency between the maximum frequency of the original signal and the minimum frequency of the aliasing noises. It is thus possible to set the same cut-off frequency of the low-pass filter for the PCM data corresponding to all the channels.

In accordance with the present invention, further, the PCM data for the respective channels are summed to convert them into an analog signal which in turn is applied to a low-pass filter, rather than the provision of separate low-pass filters for the respective channels. Accordingly, the aliasing noises included in the PCM data for the respective channels can be removed by means of a common low-pass filter. This allows the entire construction of the system to be manufactured more simply and less costly.

In accordance with the present invention, further, the PCM data will not be limited to those read out from the waveform memory through each of the channels. For example, different PCM data can be read out from the waveform memory through the same channel. Thus, the signal synthesizing system according to the present invention is very high in universality for each channel. Even if PCM data in excess of the capacity corresponding to the number of channels are stored in the waveform memory, any PCM data combination can be optionally read out from the waveform memory through each of the channels to synthesize them into an analog signal.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a circuit diagram of a preferred embodiment of an analog signal synthesizing system in PCM constructed according to the present invention.

Figure 2 illustrates an example of oversampling; Figure 2A shows the spectra of frequency in PCM data corresponding to each of the channels prior to the oversampling; and Figure 2B shows the spectra of frequency after the oversampling.

Figure 3 is a block diagram of the entire construction of an analog signal synthesizing system to which the present invention is applied.

Figure 4 illustrates the memory map in the work memory shown in Figure 3.

Figures 5 and 6 illustrate given areas in the memory map shown in Figure 4B.

Figure 7 illustrates a linear interpolation.

Figure 8 illustrates the conversion of an analog signal into a PCM signal; Figure 8A shows the sampling of the analog signal with a predetermined frequency; and Figure 8B illustrates the sampled PAM wave.

Figure 9 shows the spectra of frequency in the PCM data which are sampled with given sampling frequencies f_s.

Figure 10 is a block diagram of a circuit usable in the conventional analog signal synthesizing system in PCM.

Figure 11 illustrates cut-off frequencies for the PCM data; Figure 11A shows the sampling frequency set one-half the cut-off frequency; and Figure 11B shows the sampling frequency set one-half higher than the cut-off frequency.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

The present invention will now be described in connection with a preferred embodiment thereof illustrated in the drawings.

Referring now to Figure 3, there is shown an analog signal synthesizing system which is an preferred embodiment of the present invention.

The analog signal synthesizing system comprises a waveform memory 10 which stores a plurality of voice signals (analog signals) as PCM data which are sampled with different sampling frequencies. PCM data corresponding to plural channels are read out from the waveform memory 10 and used to produce an output synthesized analog voice signal 100R for a right-hand speaker and an output synthesized analog voice signal 100L for a left-hand speaker.

To this end, the system also comprises a work memory 20, a CPU 22 and a multi-channel programmable sound synthesizer 24.

The work memory 20 is provided with channel areas 0 - 23 addressed by addresses 0 - 17FH and interrupt areas addressed by addresses 1F8H - 1FFH, as shown in Figure 4A. The channel areas are so arranged as shown in Figure 4B while the interrupt areas are so formed as shown in Figure 4C.

The magnitudes of left- and right-hand sounds in each of the channels are written in L-volume areas and R-volume areas for that channel as shown in Figure 4B, respectively. Data representative of the musical interval of a voice outputted through that channel are written in the frequency area thereof. Such flags as shown in Figures 5 and 6 are written in the flag area. At the start and end addresses, read-start and read-end addresses for the waveform memory 10 are respectively written

in the work memory 20. These read-start and read-end addresses are used to address PCM data to be read out from the waveform memory 10 through the corresponding channel. The repeat address includes a repeat-start address written therein for repeatedly reading the addressed PCM data.

CPU 22 is adapted to perform a computation of analog signal synthesization in accordance with an operational program with the result being applied to the multi-channel programmable sound synthesizer 24 through the work memory 20.

More particularly, the CPU 22 computes data to be written in each of the areas shown in Figure 4A with the computed data being then written in the work memory 20. In addition, the CPU 22 outputs various data required to read the PCM data from the waveform memory 10 through each of the channels 0 - 23, which data are then applied to the multi-channel programmable sound synthesizer 24.

Figure 1 shows the circuit arrangement of the multi-channel programmable sound synthesizer 24.

The synthesizer 24 comprises a control circuit 30 for reading the PCM data from the waveform memory 10 for each channel, based on both the data outputted from the CPU 22 and the data written in the work memory 20; and an oversampling circuit 32 for oversampling the PCM data read out from the waveform memory 10 for each channel.

The control circuit 30 is adapted to sequentially output reading-out addresses for the PCM data from the respective channels 0 - 23 toward the waveform memory 10 using the time sharing process. Thus, the PCM data for each channel are addressed by the corresponding reading-out address and sequentially read out from the waveform memory 10. The PCM data so read are then applied to the oversampling circuit 32.

If the musical interval is set higher, the control circuit 30 will increment the PCM data reading-out address at a shortened time interval. If the musical interval is set lower, the control circuit 30 will increment the reading-out address at a prolonged time interval.

At the same time, the control circuit 30 outputs frequency data 120 representing the musical interval of the read PCM data toward the oversampling circuit 32 while outputting volume data 130R and 130L indicating the right- and left-hand volumes toward multipliers 46R and 46L, respectively.

The oversampling circuit 32 is adapted to oversample the PCM data read out from the waveform memory 10 for each channel so that the sampling frequency of the read PCM data will be shifted toward the side of higher frequency.

The oversampling may be accomplished by using any suitable process such as primary interpolation (linear interpolation), secondary interpolation, digital filtering or the like. In the illustrated embodiment, the PCM data is processed by using the linear interpolation such that they will be shifted toward the side of higher frequency.

It is now assumed that PCM data for a channel are read out from the waveform memory 10 in the order of H_{n-1}, H_n and so on as shown in Figure 7. In such a case, the m-th interpolation data H between the PCM data H_{n-1} and H_n can be determined as follows (m = 0, 1, ...):

$$H = H_{n-1} + \Delta h$$
$$= H_{n-1} + (\Delta H/\Delta T)\, m\Delta t$$
$$= H_{n-1} + (H_n - H_{n-1})\, m\Delta t/\Delta T \qquad (1)$$

where $\Delta T = T_n - T_{n-1}$; and each of T_{n-1}, T_n ... represents time at which the PCM data for each channel are outputted from the waveform memory 10.

When one of the interpolation data between the PCM data H_{n-1} and H_n is determined by such a linear interpolation equation, the sampling frequency f_s of the PCM data read out from the waveform memory 10 can be shifted toward the side of higher frequency by two times the sampling frequency, that is, up to $2f_s$. If two of the interpolation data between the PCM data are determined, the sampling frequency can substantially be increased up to three times.

As seen from Figure 1, the PCM data read out from the waveform memory 10 at this time are 12 bits, the data outputted from a subtractor 36 are 13 bits and the data outputted from a frequency data memory 40 are 8 bits. These data are computed at multiplier 38 and adder 44 for interpolation. The resulting PCM data will be extended to 16 bits with its resolution being increased.

For the linear interpolation, the oversampling circuit 32 comprises a latch circuit 34, the subtractor 36, the multiplier 38, the frequency data memory 40, a frequency data adder 42 and an adder 44.

The PCM data H_n read out from the waveform memory 10 are applied to the latch circuit 34 and the subtractor 36.

The latch circuit 34 is adapted to output the previously inputted PCM data H_{n-1} toward the subtractor 36 and adder 44. From the PCM data H_n and H_{n-1} thus inputted, the subtractor 36 computes $\Delta H = H_n - H_{n-1}$ which in turn is applied to the adder 38.

The frequency data memory 40 receives an initial value ($\Delta t/\Delta T$) from the control circuit 30 as frequency data representative of the musical interval of the PCM data. The frequency data adder 42 utilizes this initial value ($\Delta t/\Delta T$) to compute m($\Delta t/\Delta T$) which in turn is applied to the multiplier 38.

Based on the data so inputted, the multiplier 38 computes $\Delta h = (\Delta H/\Delta T)\, m\Delta t$ which in turn is applied to the adder 44.

The adder 44 totalizes the data thus inputted therein to compute the linear interpolation data H shown in the equation (1), which in turn are applied to the multipliers 46R and 46L.

Each of the multipliers 46R and 46L multiplies the PCM data from the oversampling circuit 32 by the right- or left-hand data 130R or 130L with the result being then applied to the corresponding one of right-channel and left-channel accumulators 48R and 48L.

The multi-channel programmable sound synthesizer 24 is adapted to repeat the aforementioned computation for all the channels 0 - 23 using the time sharing process. The computed data corresponding to all the channels 0 - 23 are sequentially accumulated at the accumulators 48R and 48L.

If the PCM data for the 23-th channel have been accumulated, the accumulated values in the right- and left-channel accumulators 48R and 48L are then applied sequentially to a shift register 52 through a multiplexer 50. The multiplexer 50 then converts the accumulated values for the left- and right-channels into serial data which in turn are outputted therefrom. These serial data are then applied to a D/A converter 62 through a serial-parallel converting circuit 60 shown in Figure 3.

The conversion of the accumulated values into the serial data is performed since the synthesizer 24 of the illustrated embodiment is in the form of a serial data output type one-chip element which requires the reduced number of output pins. On the contrary, if the synthesizer 24 is in the form of a parallel data output type one-chip element having the increased number of output pins, the multiplexer 50, shift register 52 and serial-parallel converting circuit 60 may be omitted.

After the accumulated values for the right- and left-channels have been inputted in the D/A converter 62 through the serial-parallel converting circuit 60, they are converted into analog signals thereat which in turn are applied to the right-and left-channel low-pass filters 64R and 64L, respectively.

Each of the low-pass filters 64R and 64L has its own cut-off frequency f_0 set based on the corresponding sampling frequency which has been shifted to the side of higher frequency. As a result, any aliasing noise included the PCM data may be removed from the analog signals inputted therein. The analog signals are then outputted therefrom through amplifiers 66R and 66L as voice signals 100R and 100L for right- and left-hand channels.

The operation of such an arrangement will be described below:

The signal synthesizing system of the illustrated embodiment is adapted to sequentially read PCM data from the waveform memory 10 for each of the channels 0 - 23 through the time sharing process.

Figure 2A shows the spectra of frequency in the PCM data read out from the waveform memory 10 for the respective channels. As described hereinbefore, the PCM data have its sampling frequency f_s which has been shifted to two times the maximum frequency of the original analog signal. Therefore, the PCM data read out from the waveform memory 10 through each of the channels will include the spectrum of the original signal shown by hatching which is created in close proximity to the spectrum of the aliasing noise. If there is not set a cut-off frequency f_c inherent in each of the channels, no aliasing noise can be removed reliably from the PCM data.

On the contrary, the analog signal synthesizing system of the present invention comprises the oversampling circuit 32 for appropriately processing the PCM data read out from the waveform memory 10 for each channel so that the sampling frequency of the PCM data for each channel will be shifted to the side of higher frequency as shown in Figure 2B.

In other words, the illustrated embodiment is adapted to determine the data H between the adjacent PCM data H_{n-1} and H_n by means of the linear interpolation, as shown in Figure 7. Therefore, the rows of data having lower sampling frequencies shown in Figure 2A can be converted into the other rows of data which are assumed that they are sampled with higher frequencies f_{DA}, such that the apparent sampling frequency f_{DA} in the data row for each channel will be forcedly shifted to the side of higher frequency.

At this time, if one of the PCM interpolated data H is determined between the adjacent PCM data, the apparent sampling frequency can be increased up to two times. If interpolated data H of n in number (n is an integer) are determined between the respective adjacent PCM data, the apparent sampling frequency f_{DA} can be increased up to $(n+1)$ times the sampling frequency f_s.

By thus shifting the apparent sampling frequency f_{DA} of the PCM data read out from the waveform memory 10 through each channel toward the side of higher frequency, the aliasing noises having lower frequencies for each channel can be forcedly shifted toward the side of higher frequency so that the spacing of frequency between the original signal and any aliasing noise will be expanded.

Even if the oversampled PCM data for each channel are synthesized by the use of the accumulators 48R and 48L, therefore, the original signal included in the synthesized PCM data will be overlapped by any aliasing noise with the spacing of frequency therebetween becoming sufficient.

In this connection, the sampling frequencies of

FIG. 1

FIG. 1A	FIG. 1B

FIG. 1A

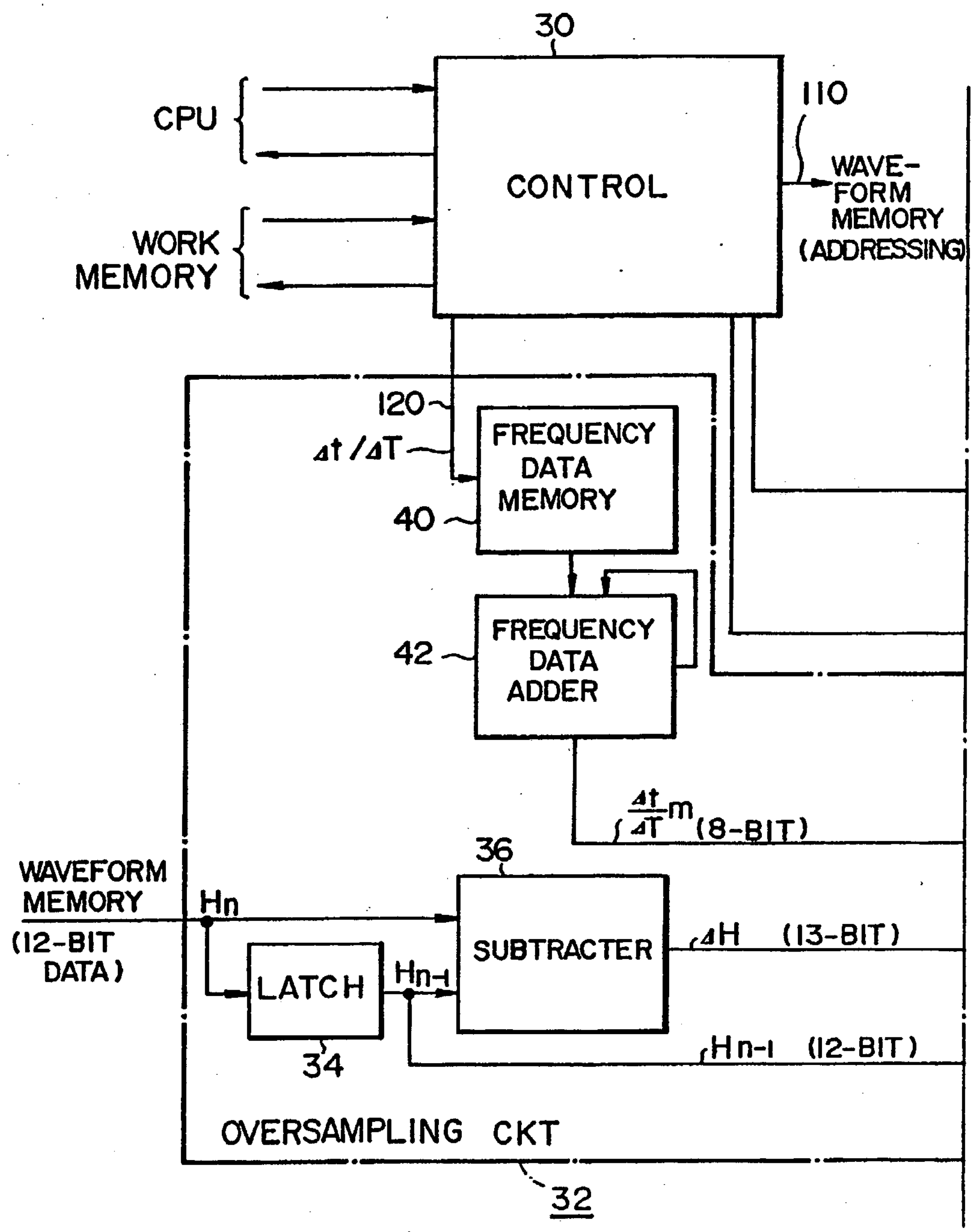
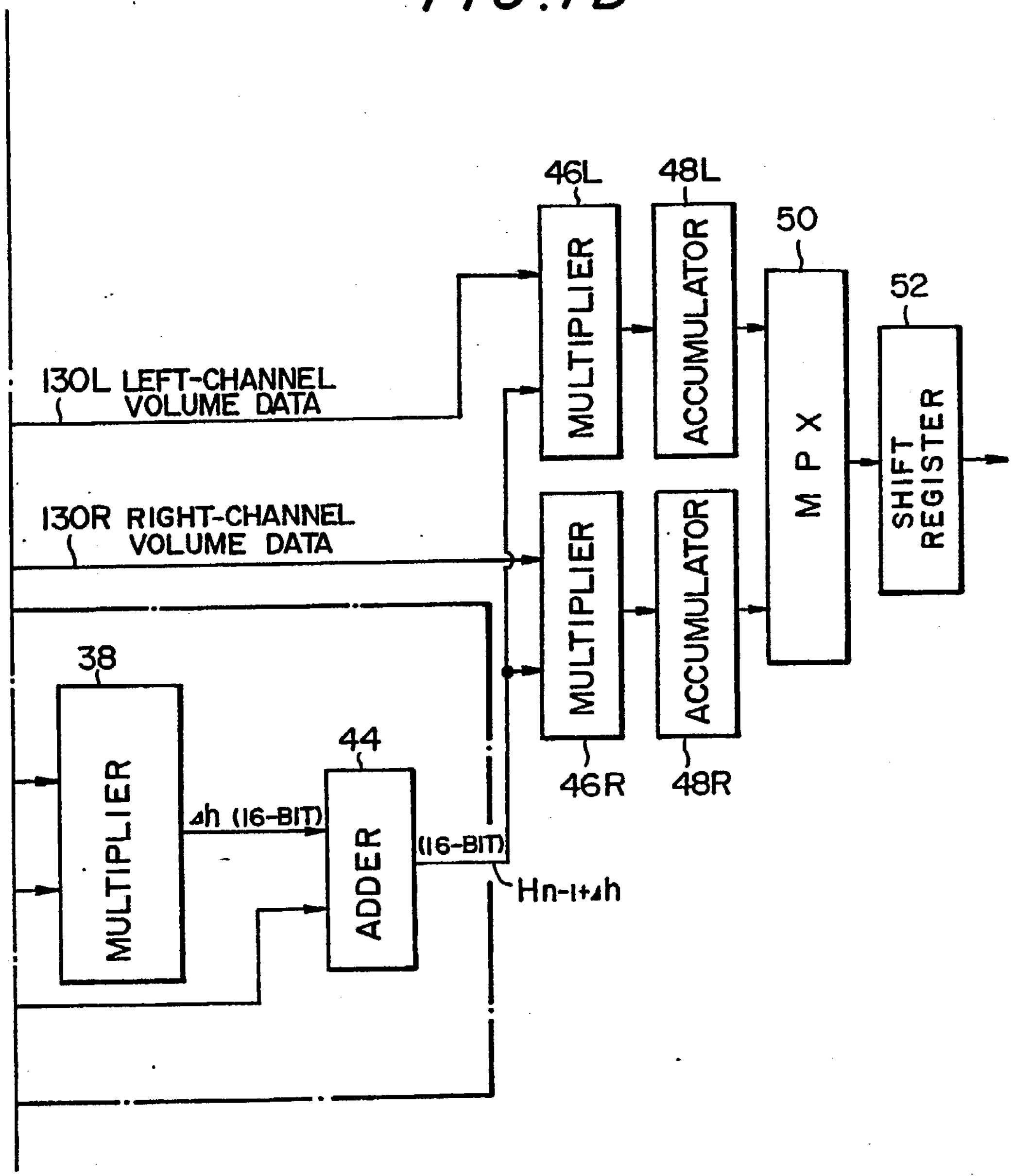


FIG. 1B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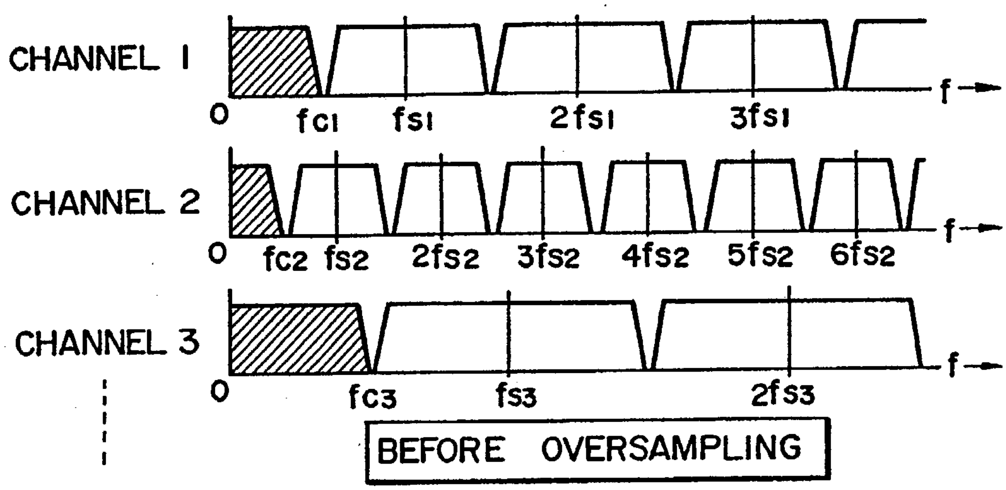
130R RIGHT-CHANNEL VOLUME DATA
MULTIPLIER
ACCUMULATOR
46R
48R
38
MULTIPLIER
44
ADDER
Δh (16-BIT)
(16-BIT)
Hn-1+Δh

FIG. 2A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CHANNEL 3
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fc3
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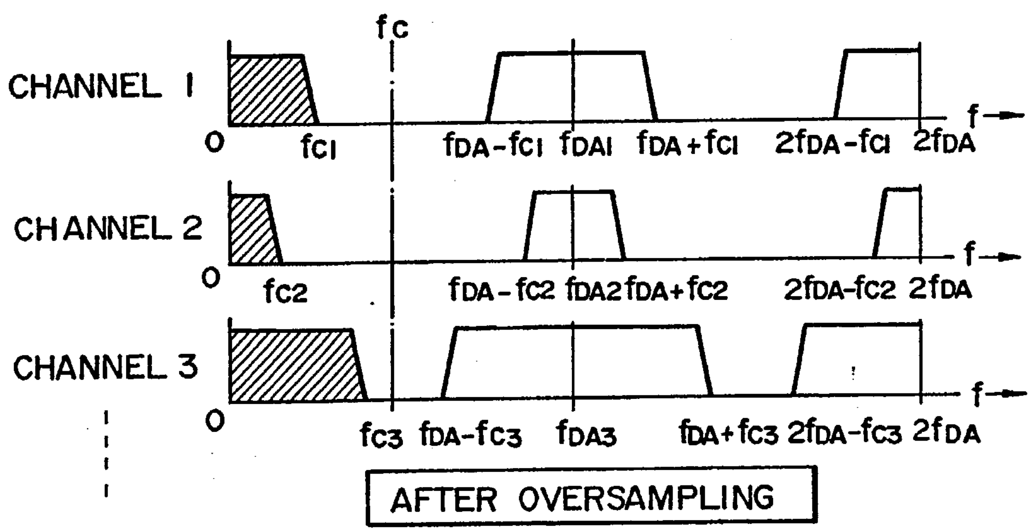
f
BEFORE OVERSAMPLING

FIG. 2B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fDA-fc1
fDA1
fDA+fc1
2fDA-fc1
2fDA
f
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fDA2
fDA+fc2
2fDA-fc2
2fDA
f
CHANNEL 3
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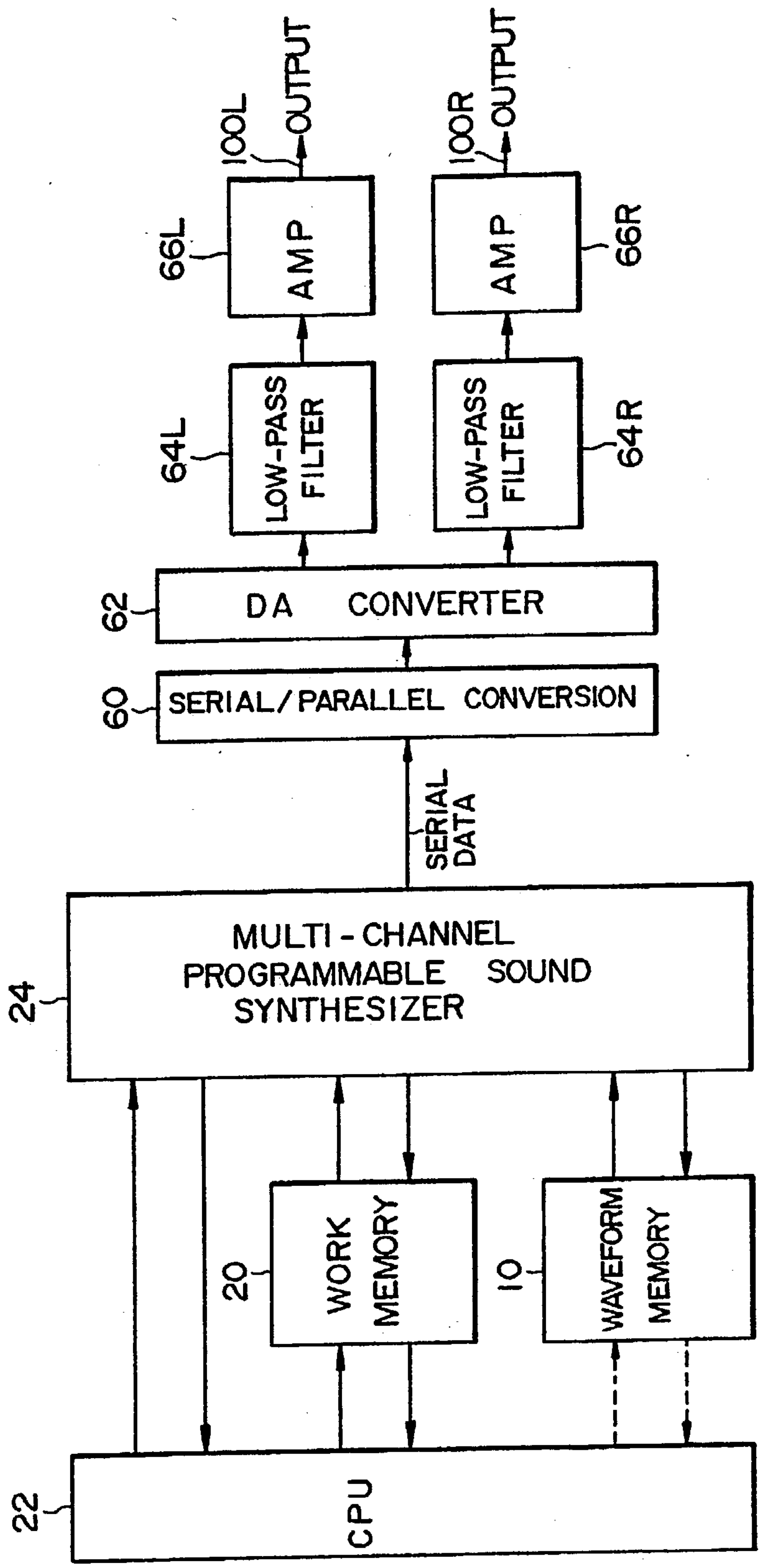
fDA+fc3
2fDA-fc3
2fDA
f
AFTER OVERSAMPLING

FIG. 3



OUTPUT
100L
OUTPUT
100R
AMP
66L
AMP
66R
LOW-PASS FILTER
64L
LOW-PASS FILTER
64R
62
DA CONVERTER
60
SERIAL/PARALLEL CONVERSION
SERIAL DATA
MULTI-CHANNEL PROGRAMMABLE SOUND SYNTHESIZER
24
WORK MEMORY
20
WAVEFORM MEMORY
10
CPU
22

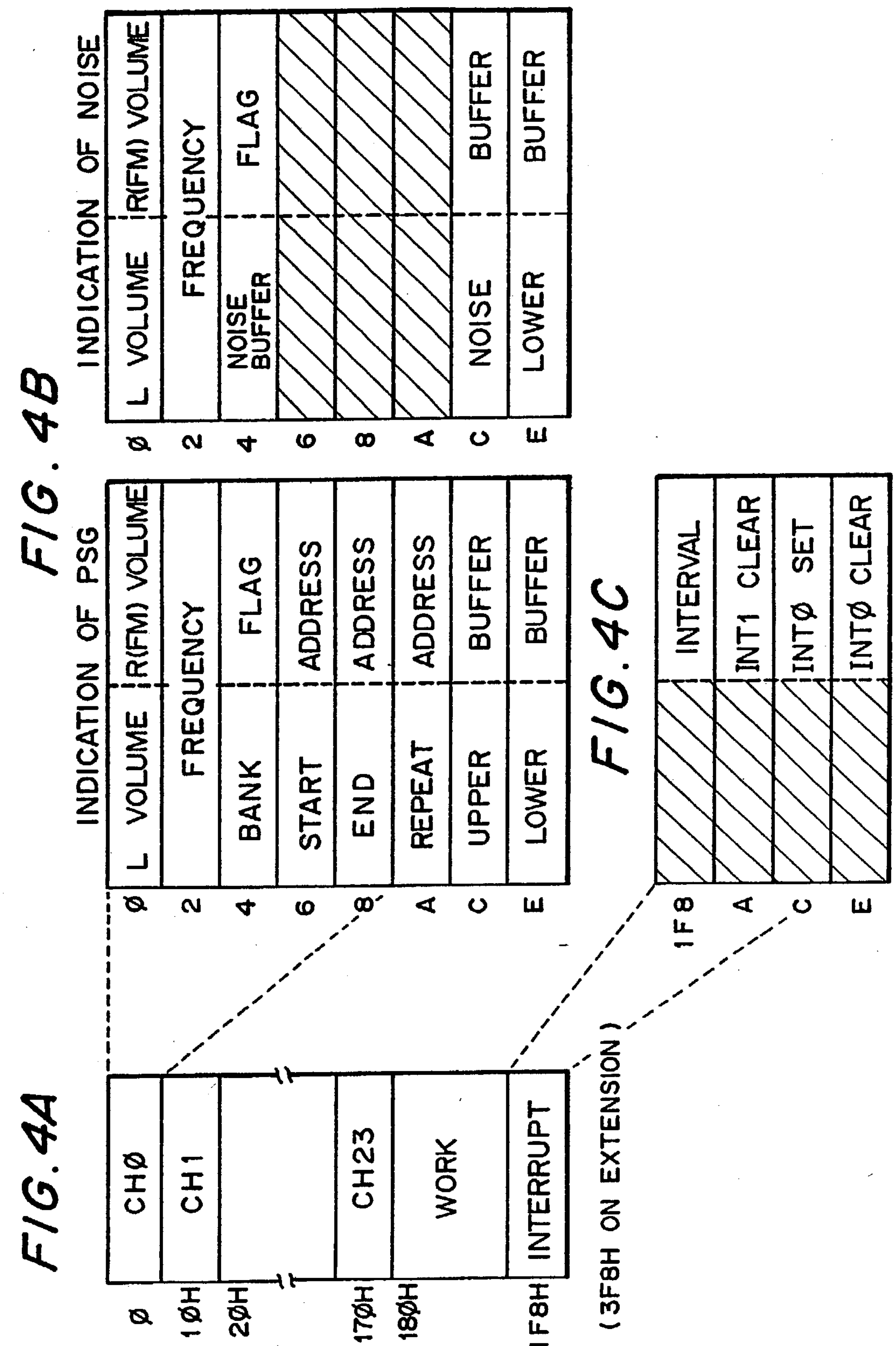


FIG. 4A

FIG. 4B

FIG. 4C

FIG. 5

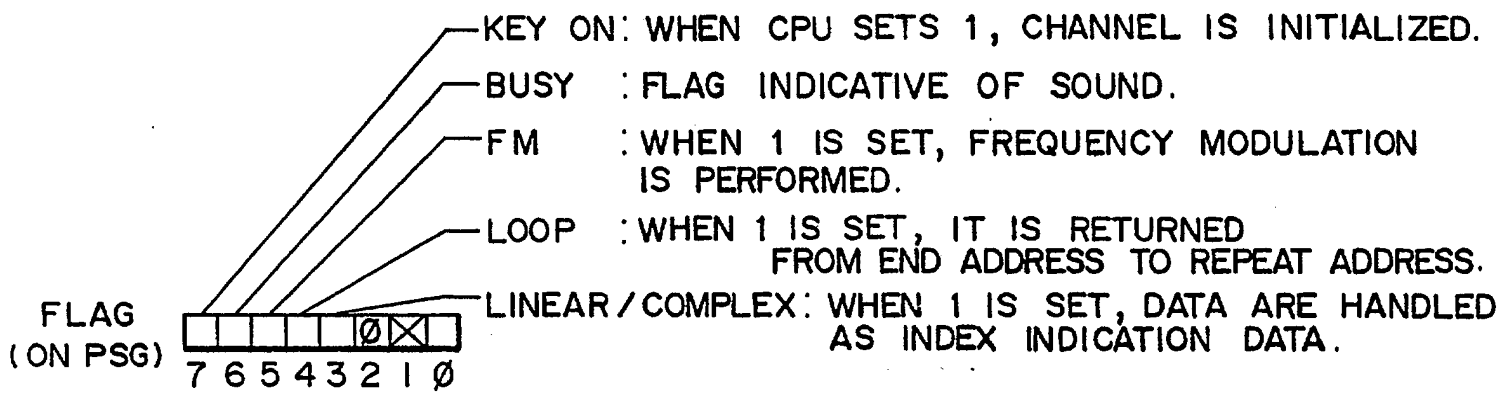

FLAG
(ON PSG)
7 6 5 4 3 2 1 Ø
Ø
KEY ON: WHEN CPU SETS 1, CHANNEL IS INITIALIZED.
BUSY : FLAG INDICATIVE OF SOUND.
FM : WHEN 1 IS SET, FREQUENCY MODULATION IS PERFORMED.
LOOP : WHEN 1 IS SET, IT IS RETURNED FROM END ADDRESS TO REPEAT ADDRESS.
LINEAR / COMPLEX: WHEN 1 IS SET, DATA ARE HANDLED AS INDEX INDICATION DATA.

FIG. 6

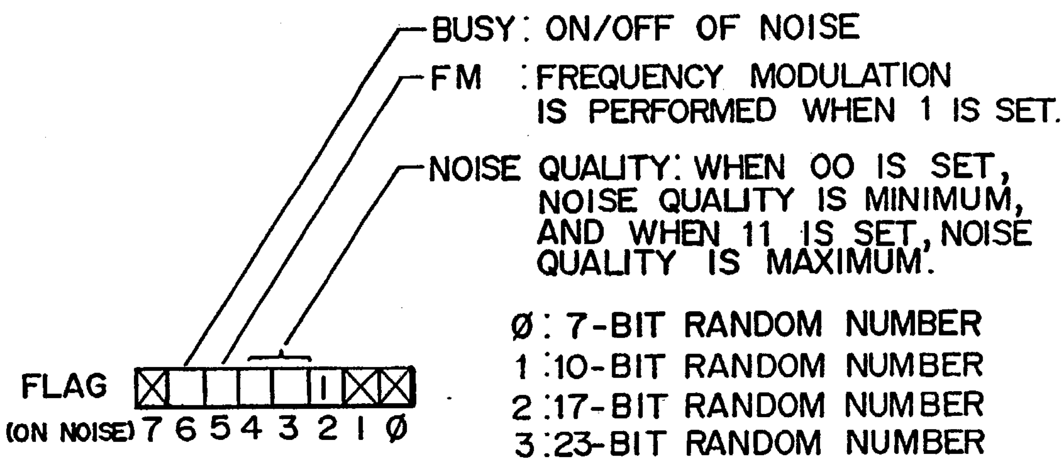

BUSY: ON/OFF OF NOISE
FM : FREQUENCY MODULATION IS PERFORMED WHEN 1 IS SET.
NOISE QUALITY: WHEN 00 IS SET, NOISE QUALITY IS MINIMUM, AND WHEN 11 IS SET, NOISE QUALITY IS MAXIMUM.
Ø: 7-BIT RANDOM NUMBER
1 :10-BIT RANDOM NUMBER
2 :17-BIT RANDOM NUMBER
3 :23-BIT RANDOM NUMBER
FLAG
(ON NOISE) 7 6 5 4 3 2 1 Ø
1

FIG. 7

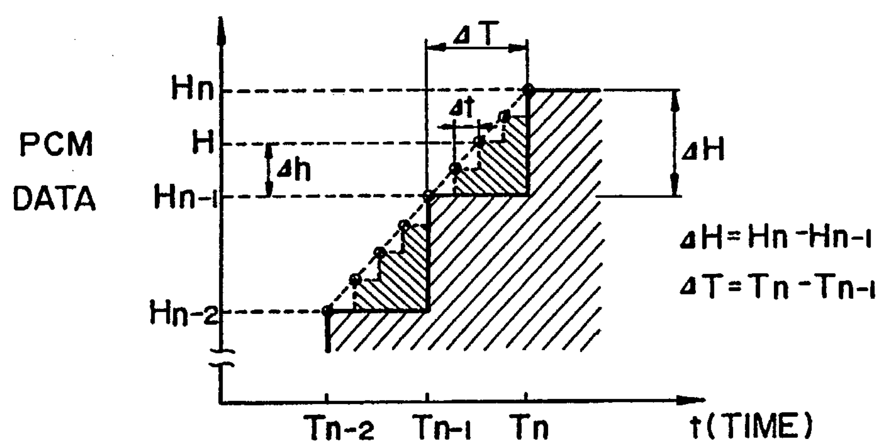

PCM
DATA
Hn
H
Hn-1
Hn-2
ΔT
Δt
Δh
ΔH
ΔH = Hn - Hn-1
ΔT = Tn - Tn-1
Tn-2
Tn-1
Tn
t(TIME)

FIG. 8A

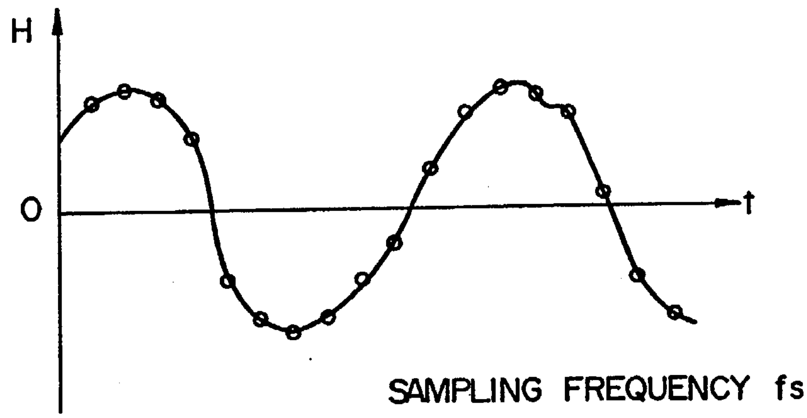


FIG. 8B

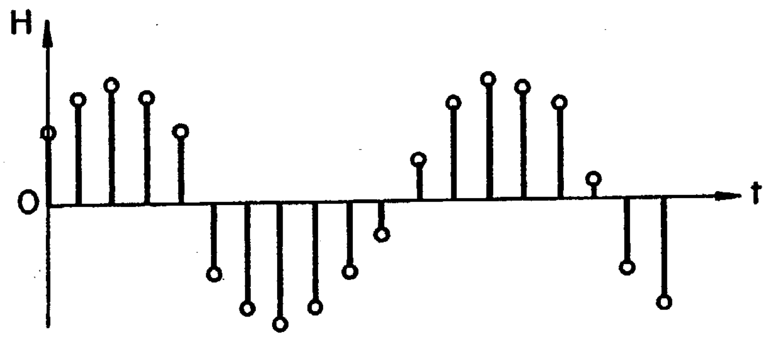
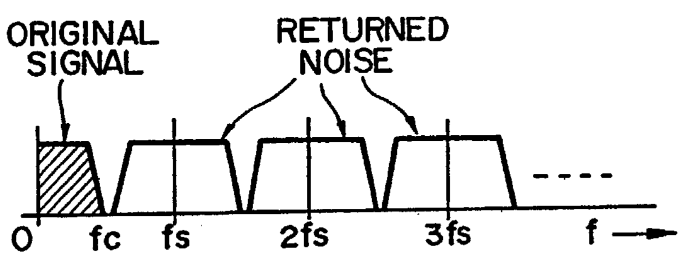


FIG. 9



fs: SAMPLING FREQUENCY

fc: ORIGINAL SIGNAL CUTTING-OFF FREQUENCY

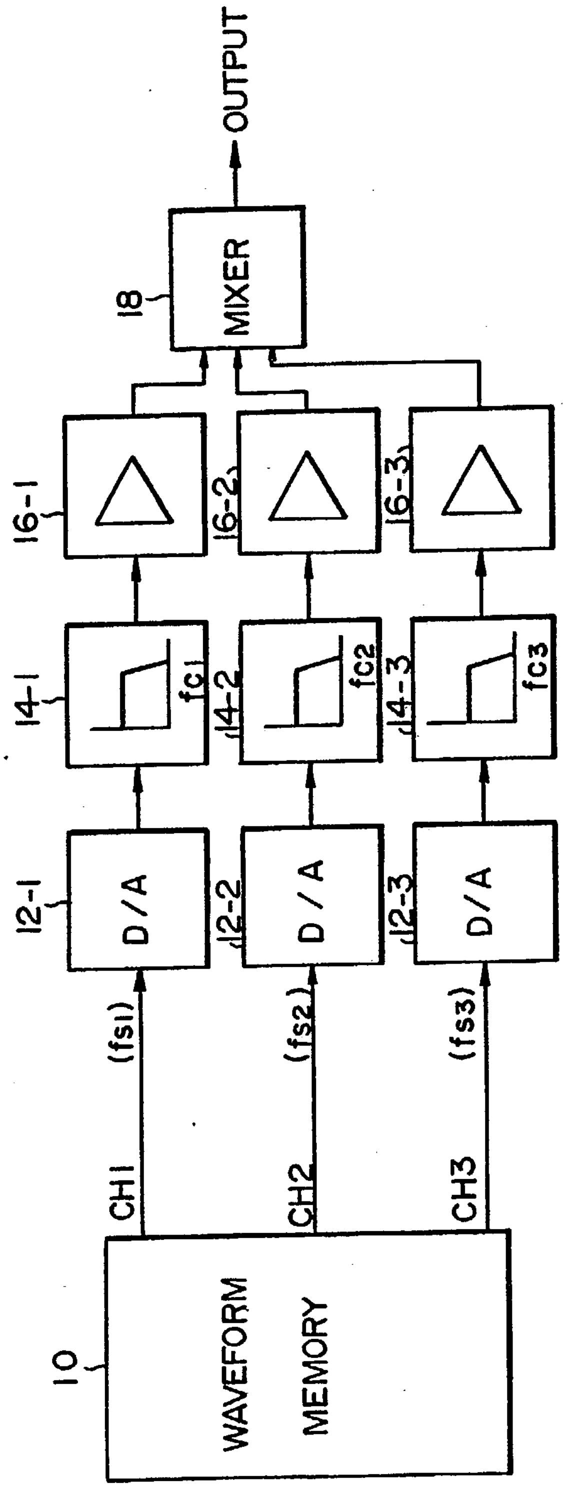
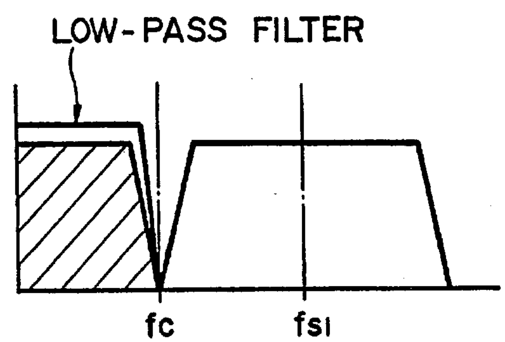

OUTPUT
18
MIXER
16-1
16-2
16-3
14-1
14-2
14-3
fc1
fc2
fc3
12-1
12-2
12-3
D / A
D / A
D / A
(fs1)
(fs2)
(fs3)
CH1
CH2
CH3
10
WAVEFORM
MEMORY

FIG. 10

FIG. 11A



$f_{s1} > f_{s2}$

FIG. 11B